DETAILED ACTION

1. This Office Action is in response to the Appeal Brief filed on 10/28/2008. Claims 1-3, 5-13, 15-17, 20, 24, and 25 remain pending. The Applicants' amendment and remarks have been carefully considered and they are found to be persuasive. Claim 1 has been amended through an Examiner's Amendment.

2. The examiner's amendment described below was discussed and authorized by the applicant in a telephone interview conducted on 01/06/2009, where claim 1 was discussed

3. All previous objections and rejections directed to the Applicant's disclosure and claims not discussed in this Office Action have been withdrawn by the Examiner.

Response to Arguments

4. Applicant's arguments (pages 4-10) filed on 10/28/2008 with regard to claims 1-3, 5-13, 15-17, 20, 24, and 25 have been fully considered and they have been found to be persuasive.

EXAMINER'S AMENDMENT

5. An examiner's amendment to the record appears below. Should the changes and/or additions be unacceptable to applicant, an amendment may be filed as provided by 37 CFR 1.312. To ensure consideration of such an amendment, it MUST be submitted no later than the payment of the issue fee.

Authorization for this examiner's amendment was given in a telephone interview with Ted Magee on 01/06/2009. The substance of the interview discussed possible amendments to overcome 35 USC 101 issues for claim 1. The following amendment below was provided by the Applicant's representative.

The application has been amended as follows:

Claims: **Replace Claim 1** from " A method of identifying an estimate for a noise-reduced value representing a portion of a noise-reduced speech signal, the method comprising: decomposing each frame of a noisy speech signal into a harmonic component for the frame and a random component for the frame; for each frame, determining a separate scaling parameter for the frame for at least the harmonic component wherein determining a scaling parameter for each frame of the harmonic component comprises determining a ratio of an energy of the harmonic component in the frame without the random component of the frame to an energy of the frame of the noisy speech signal; for each frame, multiplying the harmonic component of the frame by the scaling parameter of the frame for the harmonic component to form a scaled harmonic component for the frame; for each frame, multiplying the random component of the frame by a fixed scaling parameter for the random component, wherein the fixed scaling parameter is the same for all frames and is less than one to form a scaled random component for the frame; and for each frame, summing the scaled harmonic component for the frame and the scaled random component for the frame to form the noise-reduced value representing a frame of a noise-reduced speech signal wherein the frame of the noise-reduced speech signal has reduced noise relative to the frame of the

noisy speech signal" **to** --A method of identifying an estimate for a noise-reduced value representing a portion of a noise-reduced speech signal, the method comprising: receiving a noisy speech signal; a processor decomposing each frame of a noisy speech signal into a harmonic component for the frame and a random component for the frame; for each frame, the processor determining a separate scaling parameter for the frame for at least the harmonic component wherein determining a scaling parameter for each frame of the harmonic component comprises determining a ratio of an energy of the harmonic component in the frame without the random component of the frame to an energy of the frame of the noisy speech signal; for each frame, the processor multiplying the harmonic component of the frame by the scaling parameter of the frame for the harmonic component to form a scaled harmonic component for the frame; for each frame, the processor multiplying the random component of the frame by a fixed scaling parameter for the random component, wherein the fixed scaling parameter is the same for all frames and is less than one to form a scaled random component for the frame; and for each frame, the processor summing the scaled harmonic component for the frame and the scaled random component for the frame to produce the noise-reduced value representing a frame of a noise-reduced speech signal wherein the frame of the noise-reduced speech signal has reduced noise relative to the frame of the noisy speech signal.--.

Reasons for Allowance

6. Claims 1-3, 5-13, 15-17, 20, 24, and 25 are allowed

7. The following is an examiner's statement of reasons for allowance:

The closest prior arts of record, Laroche ("HMM: A Simple Efficient Harmonic and Noise Model for Speech", 1993) teaches the decomposing of frames into harmonic and random components (see page 1, left column, sect. 1, line 2-3). Further, Laroche teaches the summing of the harmonic and random components (see page 3, right column, sect. 4, lines 6-8). However, Laroche does not teach or suggest "determining a scaling parameter for each frame of harmonic component comprises determining a ratio of an energy of the harmonic component without the random component to an energy of the of the frame of the noisy speech signal" and "multiplying the random component …a fixed scaling parameter… is the same for all frames and is less than one…."

Rao Gadde *et al.* (US 7,120,580) is cited to disclose the multiplying the harmonic component by the scaling parameter for the harmonic component to form a scaled harmonic component (see Figure 4, multiplier 430 multiplies model weight to speech model 320); multiplying the random component by a scaling parameter for the random component to form a scaled random component (see Figure 4, multiplier 430 multiplies noise weight to the noise model 410) (e.g. The two scaling parameters or weights for the speech and noise are different, where the noise weight is 1-W and the speech weight is W). However, Rao Gadde does not teach or suggest that "for each frame, determining a ratio of an energy of the harmonic component without the random component to an energy of the of the frame of the noisy speech signal" and "for each frame, multiplying the random component …a fixed scaling parameter… is the same for all frames and is less than one…."

Rezayee (“An Adaptive KLT Approach for Speech Enhancement" is cited to disclose the harmonic scaling parameter being a ratio of an energy of the harmonic component without the random component of the frame to an energy of the noisy speech signal (see Table II, equation for gi(n)). Furthermore, Gao (US 2002/0035470) is cited to disclose determining a scaling parameter for each frame (see [0053], speech detected for each frame and a corresponding gain computed) and the scaling parameter for the random component being fixed and less than one (see [0053] and [0054]) (e.g. The scaling factor is fixed since the NSR is 1, for noise frames (random) and C is preselected and stored, where C is 0.4-0.6. Using the equation for the gain. Gf=1-(0.4:0.6), which is less then 1). However, neither Rezayee nor Gao, in combination with Laroche and Rao Gadde suggest or teach the limitation as mentioned in the above paragraphs of “for each frame, determining a ratio of an energy of the harmonic component without the random component to an energy of the of the frame of the noisy speech signal” and "for each frame, multiplying the random component …a fixed scaling parameter… is the same for all frames and is less than one….”

Thus, independent claims 1 and 13 are allowable over the prior art of record because the cited prior art alone or in combination, does not fairly suggest or disclose the claimed features, in combination, which have been mentioned above in the prior art of record.

Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably

accompany the issue fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

Conclusion

8. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Chan et al. (5,812,970) is cited to disclose noise reduction in subbands of speech signal. US Anderson et al. (US 6,453,285) is cited to disclose speech activity detection used in noise reduction. Balan et al. (US 7,110,944) is cited to disclose noise filtering of mixed sound signal filtering. Liu et al. (US 7,447,630) is cited to disclose estimation of clean speech based on decomposed signal. Visser et al. (US 7,464,029) is cited to disclose separation of speech signals in noisy environment for quality improvement. Burnett et al. (US 2002/0039425) is cited to disclose removal of noise form human speech using transfer functions. Choi (US 2008/0167863) is cited to disclose power adjustment on the basis of noise, voiced, and unvoiced signal.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to PARAS SHAH whose telephone number is (571)270-1650. The examiner can normally be reached on MON.-THURS. 7:00a.m.-4:00p.m. EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached on (571)272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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/P. S./
Examiner, Art Unit 2626

01/07/2009

/Patrick N. Edouard/

Supervisory Patent Examiner, Art Unit 2626